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INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 :
H03H 17/02, H03G 9/00

A1

(11) International Publication Number: **WO 95/25382**

(43) International Publication Date: 21 September 1995 (21.09.95)

(21) International Application Number: PCT/GB95/00602

(22) International Filing Date: 17 March 1995 (17.03.95)

(30) Priority Data:
9405211.5 17 March 1994 (17.03.94) GB

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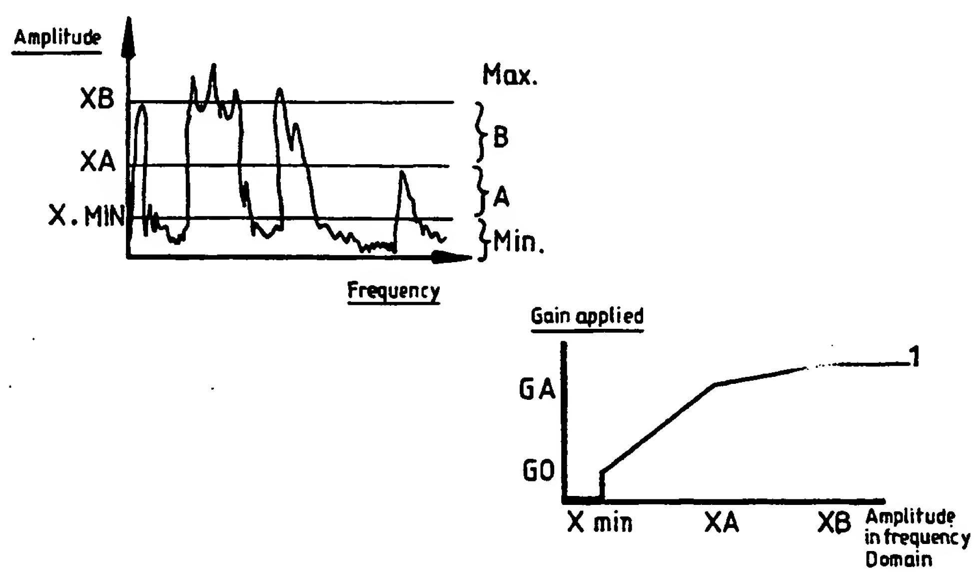
(81) Designated States: JP, NO, US, European patent (AT, BE, CH, DE, DK, ES, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).

Published
With international search report.
Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.

(54) Title: NOISE CANCELLATION SYSTEM AND METHOD



(57) Abstract

A system and method for cancelling noise in an analogue signal. The analogue signal is received and converted into a digital signal using an analogue to digital converter. The digitised signal is subsequently transformed into the frequency domain using a fast Fourier transform method. Frequency components of the frequency domain signal having an amplitude less than a predetermined threshold level are subsequently attenuated, and the filtered signal is reconverted into the time domain using an inverse fast Fourier transform. The filtered time domain signal is then reconverted into an analogue signal using a digital to analogue converter.

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NOISE CANCELLATION SYSTEM AND METHOD

The present invention relates to a system and method for reducing the noise in a signal.

It is often the case that transmitted electrical signals conveying information are contaminated by noise which may be manmade or which may originate from the natural environment. This noise can introduce errors and/or distortions into the information being conveyed which are particularly significant when the signal to noise ratio is low.

The problem of noise is particularly significant with modern telephone headsets which do not use a "boom" microphone but which instead have a single earpiece which contains both a microphone and earphone. Because of the relative remoteness of the microphone from the operator's mouth, extraneous background noise, for example from neighbouring operators, can be significant.

A number of different methods have been employed in the past to reduce the level of background noise in electrical signals. These methods include frequency shaping and voice switching (US 4,879,746) which attenuates any signals below a certain threshold, and conditions the signals above a certain threshold by using frequency shaping or adjusting the output signal amplitudes. However, in general these methods have proved unsuitable for "all-in-the-ear" type headsets.

It is an object of the present invention to overcome or at least mitigate the disadvantages of existing noise cancellation systems and methods.

According to a first aspect of the present invention there is provided a method of reducing noise in a signal containing an information component and a noise component and comprising converting the signal from the time domain to the frequency domain, defining at least two amplitude bands in the frequency domain and assigning each frequency component into one of the bands, attenuating the frequency components in the lower band(s) relative to those in the higher band(s), and reconverting the

resulting frequency domain signal into the time domain.

The present invention results from the realisation that when a speech signal is viewed in the frequency domain, the noise present tends to fill the space in the frequency domain which contains little or no information relating to speech. Attenuating the noise-related frequency components greatly reduces the total noise power present in the signal because the noise is wideband.

In a preferred embodiment of the present invention there are defined at least three, and more preferably four amplitude bands, the threshold amplitudes separating the bands being adaptively varied in dependence upon the power in the input signal. Frequency components having an amplitude within the lowest band are completely eliminated by multiplying with a gain value of zero whilst frequency components having an amplitude within the highest frequency band are preserved by multiplying them with a gain value of one. Frequency components having amplitudes within the intermediate band(s) are attenuated, or de-emphasised, by multiplying them with a gain value of between zero and one. For the intermediate bands, the gain value is determined by a gain function, a distinct gain function existing for each intermediate band where several exist, which defines a gain value which increases with component amplitude.

According to a second aspect of the present invention there is provided a noise reduction system comprising an input for receiving a signal containing an information component and a noise component, conversion means for converting the signal from the time domain to the frequency domain, comparator means defining at least two amplitude bands in the frequency domain and for assigning each frequency component into one of the bands, amplitude scaling means for attenuating the frequency components in the lower band(s) relative to those in the higher band(s), and reconversion means for reconverting

the resulting frequency domain signal into the time domain.

According to a further aspect of the present invention there is provided a noise cancellation system comprising a unit for receiving and transmitting a signal, the unit comprising:

analogue to digital conversion means for receiving a signal which is contaminated with noise and converting the signal into a digital signal;

digital signal processing means to perform a fast Fourier transform (FFT) operation on the digital signal and to remove any spectral line which is lower in amplitude than a certain predetermined threshold value, and finally to convert the frequency domain digital signal back to a time domain signal by doing an inverse FFT operation;

digital to analogue conversion means to convert the new digital signal to an analogue signal and to transmit the new signal;

a random access read/write memory to store digital values of incoming signals and results of operations; and

a non-volatile memory which stores the algorithm for the processor to perform the necessary function.

For a better understanding of the present invention and in order to show how the same may be carried into effect reference will now be made, by way of example, to the accompanying drawings in which:

Figure 1 shows schematically a noise cancellation system embodying the present invention;

Figure 2 is a flow diagram illustrating the main stages of the noise cancellation process in the system of Figure 1;

Figures 3a to 3f show signals present at various stages in the process of Figure 2 when a first noise cancellation algorithm is employed;

Figure 4 shows a set of amplitude bands into which frequency components are assigned according to an

enhanced noise cancellation algorithm; and

Figure 5 shows a gain weighting function applied to a frequency domain signal in the algorithm of Figure 2.

There will now be described with reference to Figures 1 to 3 a noise cancellation unit suitable for use with an all-in-the-ear headset of the type described above. The unit is preferably arranged to plug into a modified telephone and is connected by a wire to the all-in-the-ear headset.

A signal processing unit 1 receives an analogue electrical signal 2 which is typically contaminated by noise. The noise may arise, for example, from the earphone which sits beside the microphone or from neighbouring persons. The input signal is sampled at a rate of 8KHz and is transmitted to a flash type analogue to digital converter (ADC) 3 which converts the analogue signal into its digital equivalent. The digital signal is received by a digital signal processor (DSP) 4 which buffers the incoming signal in blocks of 64 points.

The software for controlling the operation of the digital signal processor is stored in a non-volatile ROM memory device 5 coupled to the DSP 4. A RAM memory device 6 is also coupled to the DSP and receives the operating code from the ROM 5 on power-up. The RAM 6 also acts as a buffer for the incoming 64 sample block.

Each block is pre-emphasised before further processing. Pre-emphasis is a standard technique for compensating for the average spectral shape of speech signals and involves subtracting the previous sample point from the current sample point.

The current 64 point block is combined with the previous 64 point block to provide an enlarged 128 sample point frame. The enlarged frame is windowed using a standard Hanning window for the purpose of increasing the frequency resolution of the subsequent transform process. The resulting windowed frame is then converted into the frequency domain using a fast Fourier transform

technique. The result of this transform is a frame in the frequency domain in which the signal is defined at a number of discrete frequencies or "bins". The frequency bins are approximately 61Hz wide.

At this stage a pre-boost algorithm is applied to the frame to compensate the signal for non-linearities in the response of the microphone, or other components of the system, and involves multiplying each sample point in the block by a constant defined in a look-up table. The look-up table may be determined empirically.

The DSP 4 then performs a weighting function on the signal to ease identification of the relevant part of the signal. Any spectral line which is lower in amplitude than a certain predetermined threshold value is removed at this stage.

Once the processing is complete the DSP 4 applies a post-boost algorithm to the frame. The purpose of this is similar to that of the pre-boost algorithm, i.e. to compensate for non-linearities in the signal. An inverse FFT is then applied to the frame to convert it into the time domain. The time domain digital signal is then converted to an analogue signal by a digital to analogue converter DAC 7.

The resulting signal is a time domain block having 128 sample points. The central 64 sample points are then extracted. In order to create an analogue output signal a de-emphasis function is applied in order to compensate for the initial pre-emphasis function. This step involves adding the present sample to the previously processed sample.

Figure 3 shows examples of typical signals at various stages of processing. Figure 3a shows a typical analogue signal input to the unit 1. Figure 3b shows the same signal after an FFT operation has been performed although in reality this signal will be in digital rather than analogue form. The x-axis in Figure 3b is a measure of frequency rather than time. Figure 3c shows the FFT

signal after the pre-boost has been performed. Figure 3d shows the after effects of removal of spectral lines which were lower than a predetermined threshold. Figure 3e shows the spectrum after post-boost has been performed on the spectrum of Figure 3d. Figure 3f shows the final signal after an inverse FFT has been performed.

As an alternative to the simple noise cancellation step described above, a more complex adaptive scaling process may be employed in which, after the pre-boost step, the frequency components are partitioned and scaled according to their amplitude. The flow diagram of Figure 2 is also applicable to this modification although the details of the 'scaling' algorithm are changed.

As illustrated in Figure 4, the frequency components are assigned to one of four amplitude bands MIN, A, B, and MAX. These bands are defined by the threshold levels X_MIN, XA and XB. Each amplitude band has a corresponding gain function which is applied to the frequency components within that amplitude band and these gain functions are shown in Figure 4.

Frequency components which have an amplitude lying within the MAX band, i.e. having an amplitude greater than XB, are assumed to correspond only to speech signals and are therefore are multiplied by a gain of one. The frequency components situated within band B are multiplied with a ramped gain such that frequency components nearer the lower end of the band are attenuated relative to frequency components near the upper end of the band. The level XB which defines the upper end of band B is not constant and is varied depending upon the total power present in the signal, i.e. XB = KB x total power, where KB is a constant less than one.

Similarly, frequency components having an amplitude located within band A are multiplied by a ramped gain function. However, the ramp of band A is significantly steeper than that of band B. Again, the level XA

defining the upper level of band A is dependent upon the total power present in the signal such that XA = XB x KA, where KA is also a constant less than 1.

The upper level of band MIN is defined by X_MIN which has a fixed value. Frequency components located within this band are multiplied by a gain of zero and are thus effectively cancelled.

The slopes of the gain ramps for bands A and B are determined by the values of G0 and GA as shown in Figure 5.

It will be appreciated by the skilled person that various modifications may be made to the above described embodiment without departing from the scope of the present invention. For example, conversion of a time domain signal into the frequency domain may be achieved by methods other than a fast fourier transform. For example, wavelet transforms may be used or any other type of frequency transform including discrete fourier transforms (DFT).

Claims

1. A method of reducing noise in a signal containing an information component and a noise component and comprising converting the signal from the time domain to the frequency domain, defining at least two frequency amplitude bands and assigning each frequency component into one of the bands, attenuating the frequency components in the lower band(s) relative to those in the higher band(s), and reconverting the resulting frequency domain signal into the time domain.

2. A method according to claim 1, wherein the step of attenuating the relatively low amplitude frequency components comprises multiplying the amplitude of each frequency component by a gain value, the gain value being determined from a gain function which is distinct for each amplitude band.

3. A method according to claim 2, wherein the gain function for the upper amplitude band is one and the gain function for the lower amplitude band is zero.

4. A method according to claim 2 or 3, wherein at least three bands are defined, the gain function for the or each central band being a ramp function which increases proportionally to frequency component amplitude.

5. A method according to any one of the preceding claims, wherein at least four bands are defined.

6. A method according to any one of the preceding claims and comprising the step of adaptively varying the threshold level(s) which defines the bands, in dependence upon the power of the input signal.

7. A method according to any one of the preceding claims, wherein the input signal is an analogue signal and the method comprises the step of converting the input signal into a digital signal prior to converting it into the frequency domain and the step of converting the output time domain signal into an analogue signal.

8. A method according to claim 7, wherein the time to

frequency domain conversion is carried out using a discrete Fourier transform and the frequency to time domain reconversion is carried out using an inverse discrete Fourier transform.

9. A method according to any one of the preceding claims and comprising the step of compensating the frequency domain signal, prior to attenuating the lower frequency components, using an empirically derived frequency shaping table to compensate for signal distortion.

10. A method according to any one of the preceding claims for use in removing background noise from an audio signal.

11. A noise reduction system comprising an input for receiving a signal containing an information component and a noise component, conversion means for converting the signal from the time domain to the frequency domain, comparator means defining at least two frequency amplitude bands and for assigning each frequency component into one of the bands, amplitude scaling means for attenuating the frequency components in the lower band(s) relative to those in the higher band(s), and reconversion means for reconverting the resulting frequency domain signal into the time domain.

12. A telephone or telephone interface comprising a noise reduction system according to claim 11.

13. A noise cancellation system comprising a unit for receiving and transmitting a signal, the unit comprising:

analogue to digital conversion means for receiving a signal which is contaminated with noise and converting the signal into a digital signal;

digital signal processing means to perform a fast Fourier transform (FFT) operation on the digital signal and to remove any spectral line which is lower in amplitude than a certain predetermined threshold value, and finally to convert the frequency domain digital signal back to a time domain signal by doing an inverse

FFT operation;

digital to analogue conversion means to convert the new digital signal to an analogue signal and to transmit the new signal;

a random access read/write memory to store digital values of incoming signals and results of operations; and

a non-volatile memory which stores the algorithm for the processor to perform the necessary function.

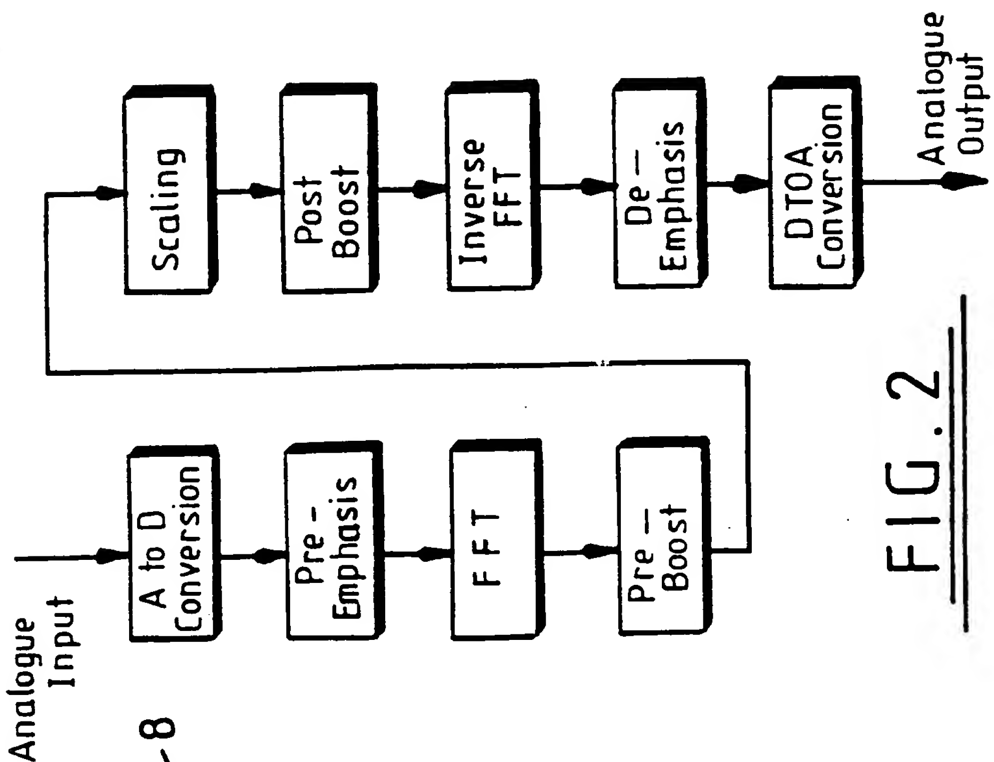
Analogue Input
A to D Conversion
Pre - Emphasis
F F T
Pre — Boost
Scaling
Post Boost
Inverse FFT
De — Emphasis
DTOA Conversion
Analogue Output

FIG. 2

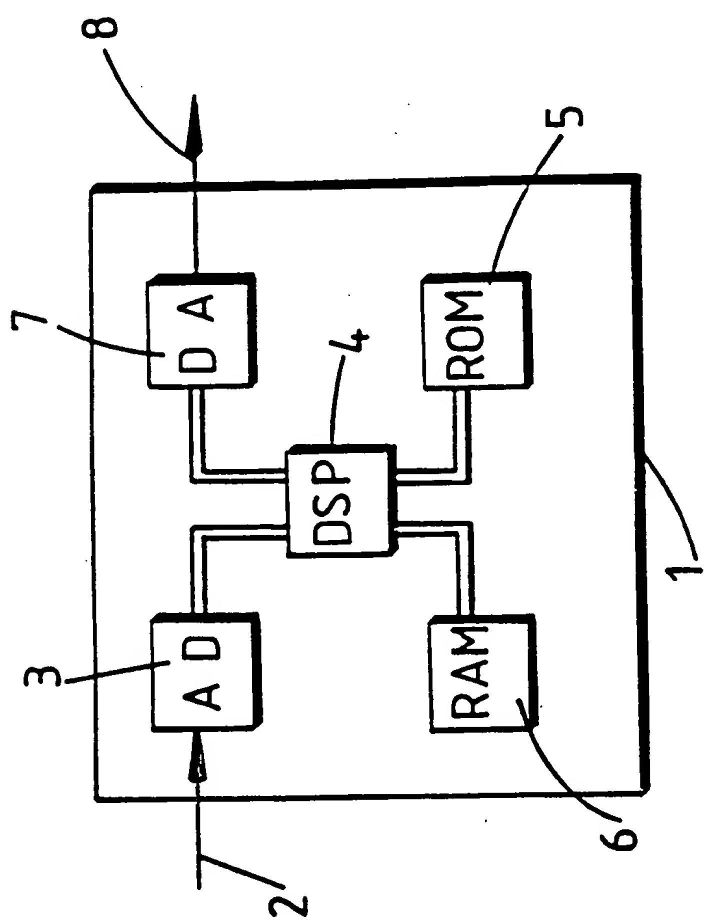
8
7
D A
4
ROM
5
DSP
3
A D
RAM
6
2
1

FIG. 1

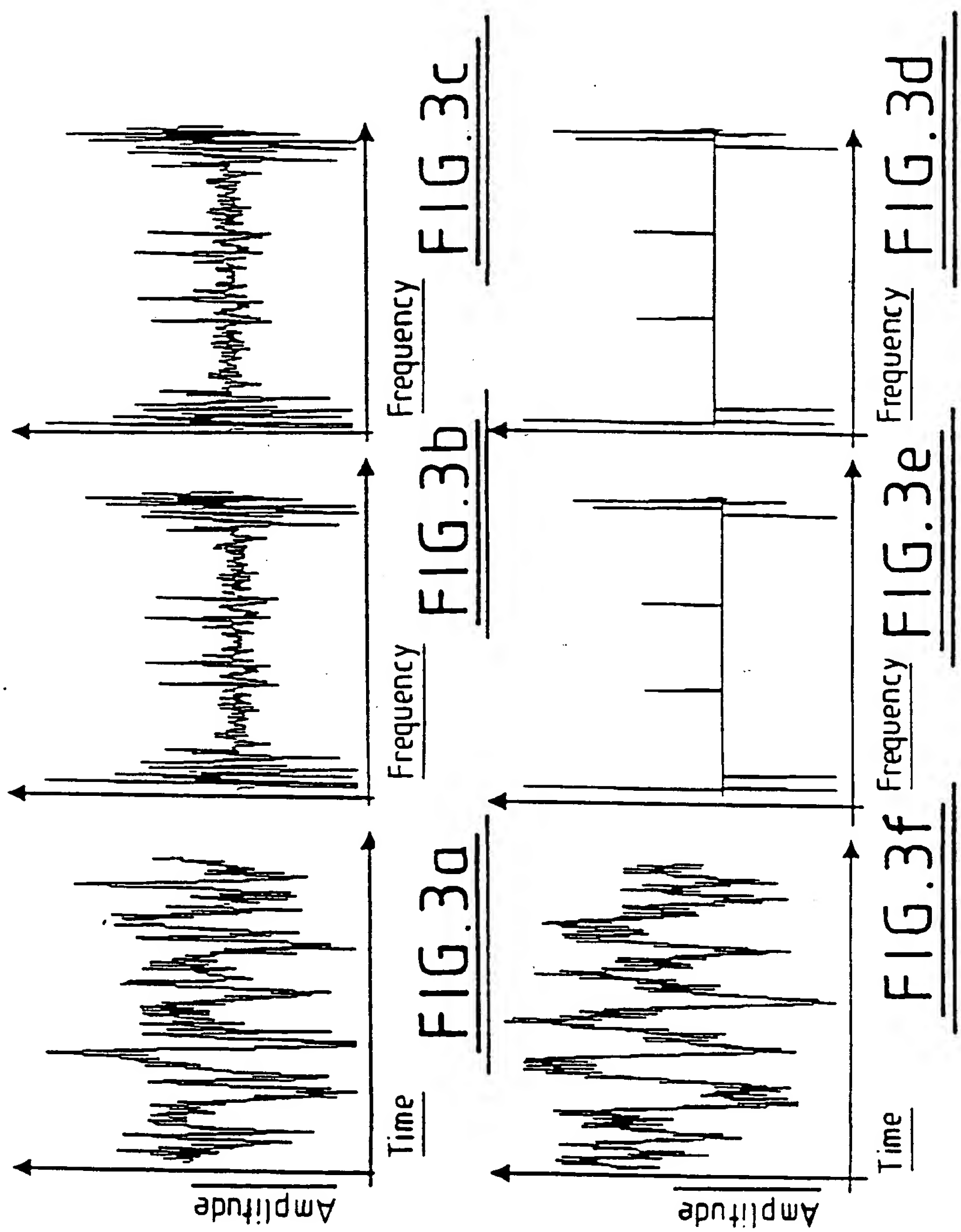

FIG.3a FIG.3b FIG.3c FIG.3d FIG.3e FIG.3f

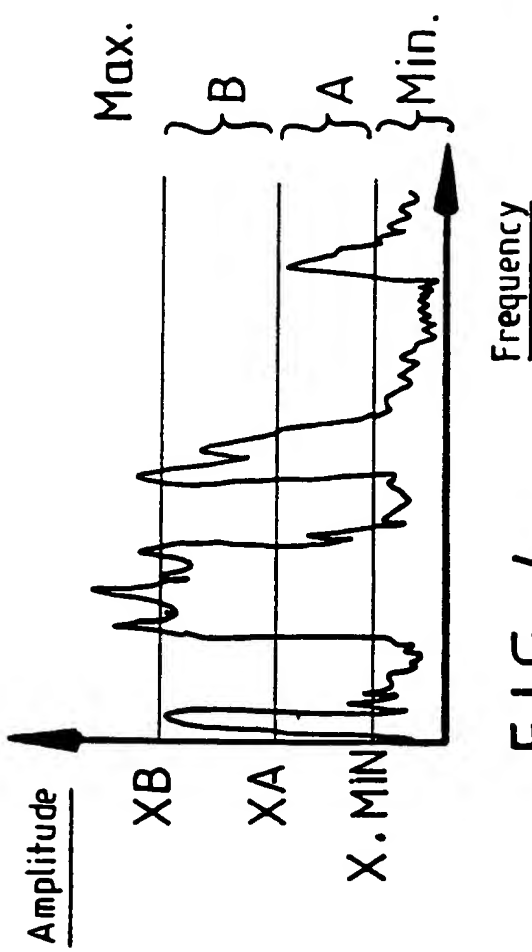
Amplitude
XB
XA
X.MIN
Max.
B
A
Min.
Frequency

FIG.4

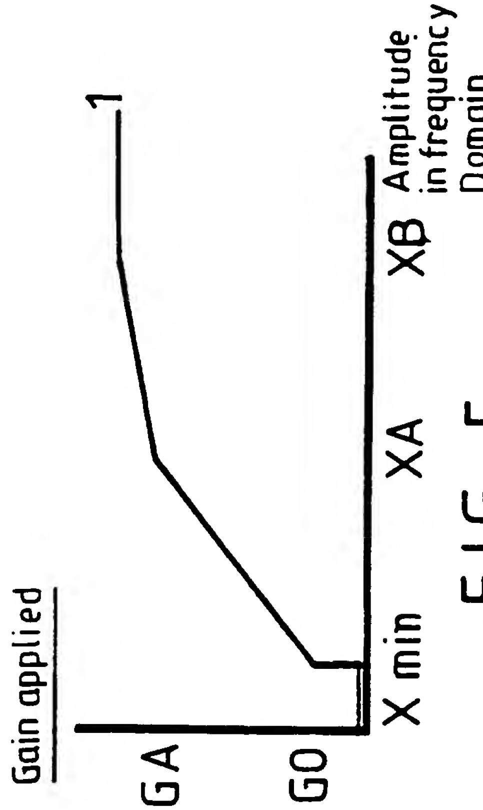
Gain applied
GA
GO
1
X min
XA
XB
Amplitude in frequency Domain

FIG.5

INTERNATIONAL SEARCH REPORT

International Application No
PCT/GB 95/00602

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H03H17/02 H03G9/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H03H H03G

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO,A,89 06877 (BRITISH TELECOM) 27 July 1989 see page 2, line 4 - page 2, line 17 see page 4, line 30 - page 5, line 30 see page 14, line 19 - page 15, line 23 ---	1,7,8, 10-12
A	EP,A,0 529 158 (LOVEJOY CONTROLS) 3 March 1993 see column 2, line 50 - column 4, line 11 -----	1

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

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Date of the actual completion of the international search	Date of mailing of the international search report
1 August 1995	07.08.95

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Information on patent family members

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Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO-A-8906877	27-07-89	CA-A- 1332626	18-10-94
		DE-D- 68913139	24-03-94
		DE-T- 68913139	23-06-94
		EP-A- 0367803	16-05-90
		GB-A- 2220330	04-01-90
		JP-T- 2503256	04-10-90
		US-A- 5133013	21-07-92
EP-A-529158	03-03-93	US-A- 5046101	03-09-91